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Docket No.: 042390.P4188C

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

In re Application of:

Graumann et al.

Assignee: Intel Corporation

Application No.: 09/747,709

Filed: 12/20/2000

For: METHOD AND APPARATUS FOR ACTIVE LATENCY CHARACTERIZATION

Examiner: Pendleton, Brian T.

Art Group: 2644

4-1-03

APPEAL BRIEF
IN SUPPORT OF APPELLANTS' APPEAL
TO THE BOARD OF PATENT APPEALS AND INTERFERENCES

Honorable Commissioner of
Patents and Trademarks
Washington, D.C. 20231

Sir:

Appellants hereby submit this Brief, in triplicate, in support of Appellants' Appeal from a final decision by the Examiner in the above-captioned case. Appellants respectfully request consideration of this Appeal by the Honorable Board of Patent Appeals and Interferences, and allowance of the claims in the above-captioned patent application.

The fee set forth in 37 CFR § 1.17(c) accompanies this Brief.

An oral hearing is NOT desired.

Please charge any additional fees to Deposit Account No. 02-2666. Also, please charge any shortages and credit any overcharges of any required fees to Deposit Account No. 02-2666.

I. REAL PARTY IN INTEREST

The invention is assigned to Intel Corporation of 2200 Mission College Boulevard, Santa Clara, California 95052

II. RELATED APPEALS AND INTERFERENCES

The subject application is a continuation of co-pending U.S. Patent Application Serial No. 08/882,381, entitled "Method and Apparatus For Active Latency Characterization," filed June 25, 1997. This parent of the subject application is currently on appeal before the Honorable Board. Appellants' Appeal Brief in the pending appeal in said parent was filed on July 22, 2002.

To the best of Appellants' knowledge, other than the pending appeal in said parent, there are no appeals or interferences related to the present appeal that will directly affect, be directly affected by, or have a bearing on the Board's decision in the subject appeal.

III. STATUS OF THE CLAIMS

Claims 1-10, 12-14, 16, 17, and 19-24 are currently pending in the subject application. These claims were finally rejected in the Final Office Action mailed December 31, 2002, and are the subject of this appeal. The Examiner confirmed the final rejection of claims 1-10, 12-14, 16, 17, and 19-24 in an Advisory Action mailed March 10, 2003.

In the Final Office Action, the Examiner raised four grounds of rejection. First, the Examiner has rejected claims 1, 12, 19, and 22 under 35 USC § 103(a) as being obvious in view of Vahatalo et al. (U.S. Patent No. 5,737,410). Second, the Examiner has rejected claims 2-4, 7-10, and 16-17 under 35 USC § 103(a) as being rendered obvious by the combination of Vahatalo et al. in view of Park et al. (U.S. Patent No. 5,410,595). Third, the Examiner has rejected claims 5-6 under 35 USC § 103(a) as being rendered obvious by Vahatalo et al. in view of Park et al. and Hollier (U.S. Patent No. 5,890,104). Fourth, the Examiner has rejected claims 13, 14, 20, 21, 23, and 24 under 35 USC § 103(a) as being obvious over Vahatalo et al. in view of Hollier. Appellants respectfully traverse each of these grounds of rejection.

IV. STATUS OF AMENDMENTS

In response to the Final Office Action, Appellants filed on February 10, 2003 an Amendment, After Final Rejection, Under Rule 116 (hereinafter, "Amendment"). In the Advisory Action, the Examiner failed to indicate whether the Amendment had been, or would be entered by the Examiner. However, in the Amendment, no claims were cancelled, amended, or added. Therefore, regardless of whether the Examiner has decided to enter the Amendment, no change in the claims would result.

In response to the Final Office Action and the Advisory Action, Appellants timely filed a Notice of Appeal on March 17, 2003. A copy of all of the claims on appeal is attached hereto as Appendix A.

V. SUMMARY OF THE INVENTION

Figure 4 of the subject application is a schematic diagram illustrating operation of a conventional speakerphone. As illustrated in Figure 4, embodiment 400 receives signals from a remote site, and these signals, when applied to speaker 410, result in acoustic output signals. Likewise, microphone 420 receives acoustic signals as input signals, and these acoustic input signals are transmitted to the remote site. For speakerphone 400 to operate in a full duplex mode, meaning in this context, that it includes the capability to both send and to receive acoustic signals at the same time, the speakerphone should employ a technique for attenuating or at least partially offsetting the acoustic coupling between speaker 410 and microphone 420. Conventionally, these techniques are referred to as acoustic echo cancellation, although perfect cancellation may not necessarily be attained. In acoustic echo cancellation, the speakerphone determines the signals originating from speaker 410 that are received by microphone 420 and attempts to attenuate or at least partially offset these acoustically coupled signals. Otherwise, a feedback loop between the remote and local site may result during full duplex operation of the speakerphone that would be undesirable. (Specification, page 4; Figure 4).

One aspect of acoustic echo cancellation relates to the timing relationship between acoustic signals produced by an acoustic signal output device, such as speaker 410, and acoustic input signals received by an acoustic signal input device, such as microphone 420. It is desirable for the acoustic echo cancellation technique to correlate, in time, audio signals or signal samples produced by the speaker with audio signals or signal samples received by the microphone. Furthermore, it is desirable to have those signal samples correlated to within a few milliseconds. One reason this is desirable in this context is because, typically, acoustic echo cancellation signal processing is implemented in the form of a digital filter of finite length. Therefore, the greater the number of taps for the filter, the more memory and processing time that is employed. Limiting the number of filter taps reduces the amount of memory employed. Thus, because the amount of memory is limited, it is desirable that the acoustic echo cancellation processing correlates signal samples between the speaker and the microphone occurring within a specified, limited time window. (Specification, page 5).

Typically, acoustic echo cancellation techniques employ a reference channel. The reference channel for the particular embodiment depicted provides a copy of the signal samples applied to the loudspeaker, although the signal samples applied are first converted to an analog signal. The signal samples are typically copied prior to this conversion. For echo cancellation to occur, it is desirable to ensure that within a time window, such as on the order of 200 milliseconds, the loudspeaker produces an audio output signal from the signal samples received by the microphone. A complicating factor with establishing this time correlation within these relatively tight time restraints is that, in a personal computer, for example, acoustic echo cancellation techniques are typically implemented using a software module or modules; however, this (these) module(s) does (do) not necessarily reside at the interface between the hardware and the software. Therefore, data buffering and other hardware/software interface issues may make it difficult to accurately and precisely establish the desired time correlation. (Specification, page 5).

In general, techniques do exist to address the problem; however, such techniques typically involve detailed knowledge regarding the hardware and/or software of the system. It would be desirable if a technique existed to characterize this latency between the speaker and the microphone that is independent of the computer hardware and software. (Specification, pages 5-6).

Figure 3 of the subject application is a block diagram illustrating an embodiment of an apparatus for active latency characterization (ALC) in accordance with the present invention. In this context, the term "active" refers to the production or creation of a waveform employed to characterize the system latency. As shown in Figure 3 of the subject application, embodiment 300 is illustrated as implemented on a personal computer (PC), although the invention is not limited in scope in this aspect. As illustrated in Figure 3 of the subject application, embodiment 300 includes a speaker 360 and a microphone 370. Likewise, this embodiment includes an acoustic echo canceller 310, a synthesizer 330, and a detector 340. Typically, these are implemented in software, although the invention is not limited in scope in this respect. As further illustrated in Figure 3 of the subject application, signals are received from a remote site. In this embodiment, these signals comprise telephony signals. It will, of course, be appreciated that the remote site and/or the peripheral devices being characterized may be coupled to the PC by a variety of communication media, including a wireless or a wired medium, for example. During this initial period, the acoustic echo canceller (AEC) 310 operates in a half-duplex mode. As illustrated, after being provided to AEC 310, signal samples are either provided along a path including synthesizer 330 and/or along path 335. For the vast majority of time, a switch 325 is selectively coupled so that the signal samples are provided along path 335. However, when it is desirable to create a predetermined waveform in the audio channel before or ahead of the audio signal output device, switch 325 is coupled to provide the signal samples along the path including synthesizer 330. This results in the creation of a predetermined waveform. The waveform created has a predetermined structure or signal signature to activate the audio signal output device to produce an audio output signal. Typically, the synthesizer is active for a

relatively short period of time, such as, on the order of 40 milliseconds. As will be discussed in more detail later, a relatively short waveform is desirable for a variety of reasons. It will, likewise, be appreciated that although switch 325 is illustrated in Figure 3, as alternatively coupling to path 335 or to synthesizer 330, this operation may also be implemented in software. Likewise, as previously indicated, the AEC, synthesizer, detector, delay and control operations illustrated in Figure 3 may all be implemented in software that operates or executes on a computer, such as a PC, although the invention is not limited in scope in this respect. (Specification, page 6; Figure 3).

As a result of operation of synthesizer 330, a waveform is produced in the form of binary digital signal samples or bits referred to here as a first signal sample stream for the waveform and provided to both speaker 360 and along the path including time delay 350 to detector 340. During this period, time delay 350 implements a delay of zero so that the signal samples traveling along this path do not experience any additional delays other than the delays associated with processing the signal samples produced. Once the signal samples for the waveform provided along this path reach detector 340, detector 340 begins to count or measure the number of signal samples received after it detects this first signal sample stream. In addition to the path of the first stream of signal samples to detector 340, as previously described, a second stream of signal samples for the waveform is also provided along a path to speaker 360. For this second stream, after digital to analog conversion, the analog signal produced is applied to speaker 360 and an audio output signal is produced. The speaker is acoustically coupled to the microphone. Therefore, an audio input signal corresponding to the audio output signal is applied to microphone 370. An acoustically coupled version of the audio output signal is then provided along a path from microphone 370 to detector 340. The second path, as illustrated in Figure 3 of the subject application, is parallel to the path including delay 350. In this context, the term "parallel" refers to the characteristic that those two paths do not intersect between the locations at which they begin and end. Detector 340, in this embodiment, counts or measures the number of signal samples received between the two detectors. It is, of course, appreciated that in an

alternative embodiment, the second signal stream may be detected first. (Specification, pages 6-7; Figure 3).

As previously described, detector 340 measures the latency or time delay between the two streams of signal samples as measured, in this embodiment, on a sample-by-sample basis. Once detector 340 has a measurement of latency, it calibrates delay 350 with at least approximately that latency and sets the AEC to full duplex operation. Thus, due to the setting of delay 350, the two paths to detector 340 are now time correlated. Therefore, detector 340 may be removed from the loop, and signal samples may be provided to AEC 310. (Specification, page 7; Figure 3).

Figure 1 of the subject application illustrates an embodiment of a synthesizer for an apparatus for active latency characterization in accordance with the present invention. To create a predetermined waveform in the audio channel, signals received from the remote site are "zeroed" in this embodiment, such as illustrated in Figure 2 of the subject application, by multiplier 110. It will, of course, be appreciated that the predetermined waveform may be super-positioned with the signals received from the remote site in an alternative embodiment. However, in this embodiment, it is considered desirable to zero the signals received from the remote site instead of super-positioning them with the waveform to reduce the possibility of corruption of the predetermined waveform due to noise, such as speech, for example. In this embodiment, the duration of the waveform may be kept sufficiently short so that substantially no speech transmitted from the remote site is rendered unintelligible. (Specification, pages 7-8; Figures 1 and 2).

Next, as illustrated in Figure 1 of the subject application, in this embodiment, a sine wave is frequency modulated. In this embodiment, this frequency modulation may be carried out in the manner described in, e.g., page 8 of the Specification. For example, in this embodiment, sine wave modulation may be implemented using a table of sine wave binary digital signal samples for a one Hertz sine wave. To produce the desired frequency modulation, the PC operates to step through the table at varying speeds. Likewise, although not illustrated in Figure 1, interpolation

may be employed in a floating point format between sine wave table signal values to provide greater precision. In this embodiment, after modulation and interpolation, resulting values of the sine wave are then multiplied by a fixed gain, such as on the order of 2^{10} and these values are provided to a digital to analog converter. The resulting analog signal is then applied to the local speaker. (Specification, pages 8-9; Figure 1).

Figure 2 of the subject application illustrates an embodiment of a detector for an apparatus for active latency characterization in accordance with the present invention. As illustrated in Figure 2, two parallel paths employing substantially the same processing are used, one for the reference channel, and the other for the local channel. In this context, the local channel refers to the path of the signals or signal samples that includes acoustic coupling between the audio signal output device and the audio signal input device. Likewise, the reference channel refers to the path of replicated signals or signal samples to be provided to the AEC through an adjustable time delay, such as, delay 350 in Figure 3 of the subject application. In this embodiment, a predetermined waveform is created in the audio channel. Thus, referring to Figure 3 of the subject application, a second stream of signal samples and a first stream of signal samples for this waveform are respectively provided along the path between speaker 360 and microphone 370 (local channel) in which acoustic coupling between these two devices occurs and, likewise, along the path including delay 350 (reference channel). It will be understood that the terms "first" and "second" do not imply any temporal relationship between the signal sample streams. (Specification, page 9; Figures 2 and 3).

In this embodiment, as illustrated in Figure 2 of the subject application, both paths include a filter, an integrator, and a threshold comparator. One reason that substantially the same processing is employed along both paths is so that when comparing the time between detections for signal streams traveling along each respective path, the time associated with processing the signal samples along the respective paths does not affect the latency determination. The signal processing employed in this embodiment implements a matched filter for the predetermined waveform and threshold trigger. Skew adjuster 280 detects a spike waveform based on the

output signal samples of the matched filter, and once it detects the desired spike, signal samples received after the spike are counted. Skew adjuster 280 eventually also detects a spike waveform for the signal samples associated with the audio signal that traveled along the path between speaker 360 and microphone 370. Skew adjuster 280, therefore, counts the number of signal samples received between these two detections and this provides an indication of the number of signal samples to delay the reference channel signal samples for the desired time correlation to occur between the reference channel and the local channel. AEC 310 may then employ the signal samples from the reference channel to perform its echo cancellation operation. One aspect of the embodiment previously described relates to maintaining a real-time relationship between the two signal sample streams. Essentially, the signal processing previously described establishes a real-time relationship between the two streams. (Specification, pages 10-11; Figures 2-3).

For the embodiment illustrated in Figure 2 of the subject application, substantially the same processing is applied to the reference channel path and the local channel path. One advantage of this embodiment is that a detector may compare in the time the detections produced for the two signal streams received without adjusting for the time attributable to performing the signal processing. Likewise, another advantage is that such an embodiment may be convenient to implement in software. Because substantially the same processing is employed along each path, the same or substantially the same software module(s) may be employed to implement the signal processing, thereby reducing the coding time. However, in an alternative embodiment in accordance with the invention, substantially the same processing need not be employed along each path. Furthermore, different predetermined waveforms may be applied along each path. The reason that such an approach may be employed is because, unlike the path between the speaker and the microphone, the reference channel path is not subject to the same risk of signal corruption. (Specification, page 11).

Figures 5A and 5B is a flowchart illustrating an embodiment of a method for active latency characterization in accordance with the invention. This embodiment may be performed,

for example, by the embodiment illustrated in Figure 3. Of course, the invention is not limited to this embodiment. As illustrated in Figure 5A, in 510, detection is begun, such as by detector 340 in Figure 3. In 515, a predetermined waveform is produced in the audio channel. For example, synthesizer 330 in Figure 3 may accomplish this. For the embodiment illustrated in Figure 5A, two signal sample streams for one predetermined waveform are produced and propagated along two different paths in the computer, as illustrated in 520. In a process loop including blocks 525, 530, and 520, the detector waits a predetermined time period. If no signal sample stream is detected within that period, then the speakerphone assumes a malfunction has occurred and stops. If a signal sample stream is detected, then the detector begins counting the signal samples that arrive after this first detection, as illustrated in 540. In a second process loop including blocks 555, 550, and 540, the detector waits for a second signal sample stream to be detected. If no detection occurs within that time period, again a malfunction is assumed. Conversely, if the second signal sample stream is detected with that time period, then the detector stops counting signal samples, as illustrated in 550. In 565, the time delay, such as illustrated in Figure 3, is adjusted based, at least in part, on the number of signal samples counted or measured by the detector. In 570, the detector is then removed from the signal sample loop. In 575, the speakerphone begins full duplex operation. (Specification, page 14, Figures 5A and 5B).

Figure 6 is a diagram illustrating a time delay between detections of the signal sample streams by an embodiment of an active latency characterization detector in accordance with the present invention. Figure 6 was not produced directly from actual results, but illustrates the type of results that an embodiment according to the present invention may be capable of producing. The detections in the reference channel and the local channel, respectively, are correlated in time. Curve 610 illustrates the results for the local channel in which a 0.4 threshold is employed, although alternative thresholds may also be used. Due to the path for the local channel for this embodiment, the results shown illustrate the effects of mixing the coherent waveform created with the local signal introduced between the speaker and the microphone. Likewise, for the reference channel, a 0.9 threshold is employed for this embodiment. However, in this

embodiment, the waveform is introduced in the audio channel along a path that includes the remote signal originating from the far end of the communications channel. Curve 620 illustrates the results that this may produce. The time delay between the detections illustrated provides a measure of latency. (Specification, pages 15-16; Figure 6).

VI. ISSUES PRESENTED

A. Whether claims 1, 12, 19, and 22 are patentable under 35 USC § 103(a) over Vahatalo et al.

B. Whether claims 2-4, 7-10, and 16-17 are patentable under 35 USC § 103(a) over the combination of Vahatalo et al. in view of Park et al.

C. Whether claims 5-6 are patentable under 35 USC § 103(a) over the combination of Vahatalo et al. in view of Park et al. and Hollier.

D. Whether claims 13, 14, 20, 21, 23, and 24 are patentable under 35 USC § 103(a) over the combination of Vahatalo et al. in view of Hollier.

VII. GROUPING OF CLAIMS

For each ground of rejection contested by Appellants in this appeal, the groupings of claims are as follows:

For purposes of the Examiner's rejection of claims 1, 12, 19, and 22 under 35 USC § 103(a) as being unpatentable over Vahatalo et al., claims 1, 12, 19, and 22 stand or fall together as Group I.

For purposes of the Examiner's rejection of claims 2-4, 7-10, and 16-17 under 35 USC § 103(a) as being unpatentable over the combination of Vahatalo et al. in view of Park et al., claims 2-4, 7-10, and 16-17 stand or fall together as Group II.

For purposes of the Examiner's rejection of claims 5-6 under 35 USC § 103(a) as being unpatentable over the combination of Vahatalo et al. in view of Park et al. and Hollier, claims 5-6 stand or fall together as Group III.

For purposes of the Examiner's rejection of claims 13, 14, 20, 21, 23, and 24 under 35 USC § 103(a) as being unpatentable over the combination of Vahatalo et al. in view of Hollier, claims 13, 14, 20, 21, 23, and 24 stand or fall together as Group IV.

Reasons for the separate patentability of Claim Groups I-IV are presented in the Argument section pursuant to 37 C.F.R. § 1.192(c)(5).

VIII. ARGUMENT

A. VAHATALO ET AL. DOES NOT TEACH OR SUGGEST CLAIMS 1, 12, 19, AND 22.

The Examiner has taken the position that claims 1, 12, 19, and 22 are rendered obvious by Vahatalo et al. In order to establish a *prima facie* case of obviousness:

> First, there must be some suggestion or motivation, either in the references themselves or in the knowledge generally available to one of ordinary skill in the art, to modify the reference or to combine reference teachings. Second, there must be a reasonable expectation of success. Finally, the prior art reference (or references when combined) must teach or suggest all the claim limitations. . . The teaching or suggestion to make the claimed combination and the reasonable expectation of success must both be found in the prior art, not in applicant's disclosure. *In re Vaech*, 947 F.2d 488, 20 USPQ2d 1438 (Fed. Cir. 1991). *Manual of Patent Examining Procedure* (MPEP), 8th Edition, August 2001, § 2143.

In the Final Office Action, concerning Vahatalo et al., the Examiner asserts:

> As disclosed in the abstract and column 4 line 63 - column 8 line 50 and figure 5B, Rin and Sin are two waveforms in an audio channel which are used to calculate the delay from the outgoing echo location and the returned echo. The delay is set in adjustable delay element 43. Vahatalo et al. do not disclose that the location of the echo is calculated for an audio channel in a computer. Echo

existed in acoustic environments and hybrid line environments. The method of determining the echo location would have been the same in either environment, as one of ordinary skill in the art would have known. Said method would require identifying the outgoing signal in part of the incoming signal, regardless if the signals were propagated through the air or transmission line. Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to apply the technique used for echo location in transmission lines in systems having acoustic echo, such as computer/telephone speakerphones. . . Vahatalo et al. do not disclose a machine readable storage medium which executes the method. Nonetheless, it was obvious at the time of invention to perform signal processing methods through the use of microcomputers where the method steps are instructions on machine readable storage media. The use of microcomputers adds efficiency and speed to the process. Final Office Action, pages 2-3.

However, claim 1, for example states:

A method for actively characterizing the latency of an audio channel of a computer, comprising:

creating at least two signal sample streams for a waveform in the audio channel;

detecting the presence of the first signal sample stream for said waveform and the second signal sample stream for said waveform at a point in said audio channel;

measuring the time between the detections of the signal sample streams; and

delaying at least one of the signal sample streams based, at least in part, on the time measured between the detections. (Independent claim 1).

Thus, **claim 1 clearly requires that two or more signal streams be created in the audio channel**. Contrary to the Examiner's suggestion, Rin and Sin are not signals, but instead, they are ports. This is made clear at, for example, Vahatalo et al., column 5, lines 5-10. Thus, Rin and Sin do not even qualify as two or more signal streams required in claim 1.

Furthermore, even assuming, for the sake of argument, that Rin and Sin represent signals, it is clear that Sin and Rin would then be electrical signals, not audio signals. Therefore, even assuming, for the sake of argument, that Sin and Rin were signals, they still would not be signals created in the audio channel. The hybrid in Vahatalo et al. pointed to by the Examiner is a circuit that operates on electrical signals, not audio signals, and **Vahatalo et al. clearly distinguishes between acoustic echo cancellation and electrical echo cancellation, and teaches that Vahatalo et al. is directed to electrical echo cancellation**.[1] See, e.g., Vahatalo et al., column 1, lines 27 - 65.

Indeed, the Examiner even admits that Vahatalo et al.'s disclosed hybrid is electrical, and that Vahatalo et al. does not relate to acoustic echo cancellation. For example, at pages 3-4 of the Final Office Action, the Examiner concedes that **"[T]he subject matter of Vahatalo et al. is directed to hybrid circuit echo cancellation and does not disclose a signal output and input device for acoustic echo cancellation."** Final Office Action, pages 3-4. Thus, Vahatalo et al. does not disclose or suggest the two or more signal streams created in the audio channel required in claim 1.

Additionally, Vahatalo et al. cannot be said to disclose or suggest delaying one or more of the signal streams based, at least in part, on the time measured between detections of the at least two signal streams, as is required in claim 1. Indeed, the Examiner has not cited any portion of Vahatalo et al. as disclosing or suggesting this additional limitation of claim 1. Accordingly, for this additional reason, Vahatalo et al. does not render obvious claim 1.

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[1] At page 3 of the Final Office Action, the Examiner apparently asserts that it is well known in the art that the method for determining acoustic echo is same as the method for determining electrical echo. Appellants respectfully traverse this assertion by the Examiner. Pursuant to MPEP § 2144.03, Appellants respectfully request that the Examiner either (1) supply a prior art reference or personal affidavit to support the Examiner's assertion, or (2) withdraw the Examiner's assertion and rejection based thereon.

Claim 12 is similar in some respects to claim 1. It states:

A method for actively characterizing the latency of an audio channel of a computer, comprising:

creating at least a first and a second waveform in said audio channel;

detecting the presence of the first and second waveform at a point in said audio channel;

measuring the time between the detections of the waveforms; and

delaying at least one of the waveforms, based at least in part, on the time measured between the detections. (Claim 12).

Once again, Rin and Sin in Vahatalo et al. are ports of an electrical circuit, not waveforms in an audio channel, as is required in claim 12. Additionally, Vahatalo et al. cannot be said to disclose or suggest delaying at least one such waveform based, at least in part, on the time measured between detections of two such waveforms, as is required in claim 12. Indeed, as is the case with claim 1, the Examiner has not cited any portion of Vahatalo et al. as disclosing or suggesting this additional limitation of claim 12. Accordingly, as with claim 1, Vahatalo et al. fails to render obvious claim 12.

Claim 19 requires a machine-readable storage medium that stores instructions that when executed by a computer system result in creation in an audio channel of at least two signal streams for a waveform in an audio channel. Claim 22 requires a machine-readable storage medium that stores instructions that when executed by a computer system results in creation in an audio channel of at least two signal waveforms. As stated above, Rin and Sin in Vahatalo et al. are ports of an electrical circuit, not signal streams or waveforms in an audio channel.[2] Accordingly, Vahatalo et al. does not disclose or suggest claims 19 and 22.

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[2] The Examiner apparently acknowledges at page 3 of the Final Office Action that Vahatalo et al. fails to disclose a machine-readable storage medium that stores instructions whose execution results in the operations recited in claims 19 and 22, but nonetheless, the Examiner still asserts that claims 19 and 22 are rendered obvious by Vahatalo et al. Pursuant to MPEP § 2144.03, Appellants respectfully request that the Examiner either (1) supply a prior art reference or personal affidavit suggesting the desirability of storing in a machine-readable medium instructions

Thus, in summary, Vahatalo et al. does not teach or suggest, among other things, either (1) creating two signal streams or waveforms in the audio channel, or (2) delaying one or more of the signal streams or waveforms based, at least in part, on the time measured between detections of the at least two signal streams or waveforms. Accordingly, Vahatalo et al. cannot render obvious claims 1, 12, 19, and 22. Therefore, it is respectfully submitted that the Examiner's final rejection of claims 1, 12, 19, and 22 under 35 USC § 103(a) as being rendered obvious by Vahatalo et al. is erroneous, and should be reversed.

B. VAHATALO ET AL. AND PARK ET AL., WHETHER TAKEN SINGLY OR IN COMBINATION, DO NOT RENDER OBVIOUS CLAIMS 2-4, 7-10, AND 16-17

The deficiencies of Vahatalo et al. vis-à-vis the claimed invention have been previously described in detail. Park et al. is cited by the Examiner as disclosing "an apparatus for speakerphone (acoustic) echo cancellation having a microphone 21 . . . and speaker 22 . . ." Final Office Action, p. 4. Even assuming, for the sake of argument, that Park et al. does disclose these features, given these deficiencies of Vahatalo et al. vis-à-vis the claimed invention, the Examiner's rejection of claims 2-4, 7-10, and 16-17 as being rendered obvious by the combination of Vahatalo et al. and Park is in error.

Claims 2-4 and 7-10 depend from claim 1 and include all of the limitations of claim 1. Likewise, claims 16-17 depend from claim 12 and include all of the limitations of claims 12. For at least the reasons presented above, neither Vahatalo et al., nor Park et al., nor their combination, disclose or suggest claims 1 and 12. Specifically, neither patent, either individually or in combination, teaches or suggests (1) creating two signal streams or waveforms in the audio channel, or (2) delaying one or more of the signal streams or waveforms based, at least in part, on the time measured between detections of the at least two signal streams or waveforms.

Park et al., for example, does not recognize the latency estimation issue that exists in the computer environment. Park et al. employs hardware synchronization of separate clocks, rather

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whose execution results in the operations recited in claims 19 and 22, or (2) withdraw the Examiner's assertion and

than estimating the latency. Park et al. states that "the output clock for DAC 66 is synchronized to the input clock for ADC 61." Park et al., column 5, lines 64-66. Park et al. specifically defines terms "N" and "DELAY" in connection with this clock synchronization. DELAY is defined as N1-N2, which reflects the number of signal samples between when a signal, depicted in Figure 2 of Park et al., exceeds a threshold TH1, to when it falls below a threshold TH2. Park et al., column 5, lines 52-59 and column 6, lines 3-10. It is clear that only one signal is employed in Park et al.'s disclosed arrangement to estimate a delay (although yet another difference is that this delay is not the same or similar to the delay measured in the subject application). Likewise, Vahatalo et al. does not teach or suggest creating two signal streams in the audio channel.

Accordingly, it cannot be said that claims 2-4, 7-10, and 16-17 are rendered obvious by Vahatalo et al. in view of Park et al. Therefore, it is respectfully submitted that the Examiner's final rejection of claims 2-4, 7-10, and 16-17 under 35 USC § 103(a) as being rendered obvious by Vahatalo et al. in view of Park et al. is erroneous, and should be reversed.

C. VAHATALO ET AL., PARK ET AL., AND HOLLIER, WHETHER TAKEN SINGLY OR IN COMBINATION, DO NOT RENDER OBVIOUS CLAIMS 5-6

Claims 5-6 depend from claim 1, and thus, must be read as incorporating the limitations of claim 1. Hollier fails to cure the deficiencies pointed out above regarding Vahatalo et al. and Park et al. vis-à-vis claim 1. None of these patents, either singly or in combination, teach or suggest (1) creating two signal streams in the audio channel, or (2) delaying one or more of the signal streams based, at least in part, on the time measured between detections of the at least two signal streams, as is required in Appellants' claim 1. Hollier, for example, relates to testing telecommunications equipment. Hollier has nothing to do with an audio channel or acoustic echo cancellation, and therefore, the combination proffered by the Examiner is improper.[3] It is

rejection based thereon.

[3] At page 5 of the Final Office Action, the Examiner asserts that "use of known signals to calculate . . . delay would have been more reliable, thereby prompting one of ordinary skill in the art to use them." Appellants respectfully traverse the Examiner's assertion, and pursuant to MPEP § 2144.03, Appellants respectfully request that the Examiner (1) either supply a prior art reference or personal affidavit to support the Examiner's assertion, or (2) withdraw the Examiner's assertion and rejection based thereon.

quite clear that the Examiner has engaged in hindsight to pick through isolated disclosures in a vain effort to assemble the elements of Appellants' claims.

Accordingly, it cannot be said that claims 5-6 are rendered obvious by Vahatalo et al. in view of Park et al. and Hollier. Therefore, it is respectfully submitted that the Examiner's rejection of claims 5-6 under 35 USC § 103(a) as being rendered obvious by Vahatalo et al. in view of Park et al. and Hollier is erroneous, and should be reversed.

D. VAHATALO ET AL. AND HOLLIER, WHETHER TAKEN SINGLY OR IN COMBINATION, DO NOT RENDER OBVIOUS CLAIMS 13, 14, 20, 21, 23, AND 24

Claims 13 and 14 depend from claim 12, and thus, must be read as incorporating the limitations of claim 12. Claims 20 and 21 depend from claim 19, and thus, must be read as incorporating the limitations of claim 19. Claims 23 and 24 depend from claim 22, and thus, must be read as incorporating the limitations of claim 22. Hollier fails to cure the deficiencies pointed out above regarding Vahatalo et al. vis-à-vis claims 12, 19, and 22. Neither of these patents, either alone or in combination, teaches or suggests creating two signal streams or waveforms in the audio channel, as is required in claims 12, 19, or 22. Additionally, neither of these patents, either alone or in combination, teaches or suggests delaying one or more of the waveforms based, at least in part, on the time measured between detections of the at least two waveforms, as is required in claim 12. Additionally, for the reasons stated above, the combination proffered by the Examiner is improper. This is yet further evidence that the Examiner has engaged in hindsight to pick through isolated disclosures in a vain effort to assemble the elements of Appellants' claims!

Accordingly, it cannot be said that claims 13, 14, 20, 21, 23, and 24 are rendered obvious by Vahatalo et al. in view of Hollier. Therefore, it is respectfully submitted that the Examiner's

rejection of claims 13, 14, 20, 21, 23, and 24 under 35 USC § 103(a) as being rendered obvious by Vahatalo et al. in view of Hollier is erroneous, and should be reversed.

IX. CONCLUSION

For the foregoing reasons, Appellants respectfully submit that each and every one of the final rejections made by the Examiner in the Final Office Action is erroneous. Accordingly, Appellants respectfully request that the Honorable Board of Patent Appeals and Interferences reverse the Examiner and direct that all of the currently pending claims be allowed.

Please charge any shortages and credit any overcharges to Deposit Account number 02-2666.

Respectfully submitted,

Date: 19 March 2003

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APPENDIX A: CLAIMS ON APPEAL

1. A method for actively characterizing the latency of an audio channel of a computer, comprising:

creating at least two signal sample streams for a waveform in the audio channel;

detecting the presence of the first signal sample stream for said waveform and the second signal sample stream for said waveform at a point in said audio channel;

measuring the time between the detections of the signal samples streams; and

delaying at least one of the signal sample streams based, at least in part, on the time measured between the detections.

2. The method of claim 1, wherein said audio channel includes an audio signal output device and an audio signal input device;

creating a waveform in said audio channel comprising creating a waveform in said audio channel before said audio signal output device, said waveform having a signature to activate said audio signal output device to produce an audio output signal; and

detecting the presence of a first signal sample stream for said waveform and a second signal sample stream for said waveform at a point in said audio channel comprising detecting the signal sample streams in said audio channel at a point after said audio signal input device, wherein the first signal sample stream was propagated along a reference channel path in said computer and the second signal stream was produced from said audio output signal and propagated along a local channel path in said computer.

3. The method of claim 2, wherein the audio signal output device includes at least one speaker.

4. The method of claim 2, wherein the audio signal input device includes a microphone.

5. The method of claim 2, wherein said waveform comprises a chirp waveform.

6. The method of claim 2, wherein said waveform comprises a pseudo-random sequence waveform.

7. The method of claim 2, wherein said waveform comprises a sine waveform.

8. The method of claim 2, wherein measuring the time between the detections comprises counting the number of signal samples between the detections.

9. The method of claim 1, wherein after creation, the two signal streams propagate along two different paths in said computer.

10. The method of claim 1, wherein said computer comprises a personal computer.

12. A method for actively characterizing the latency of an audio channel of a computer, comprising:

creating at least a first and a second waveform in said audio channel;

detecting the presence of the first and second waveform at a point in said audio channel;

measuring the time between the detections of the waveforms; and

delaying at least one of the waveforms, based at least in part, on the time measured between the detections.

13. The method of claim 12, wherein at least one of said waveforms comprises a chirp waveform.

14. The method of claim 12, wherein at least one of said waveforms comprises a pseudo-random sequence waveform.

16. The method of claim 12, wherein after creation, the two waveforms propagate along two different paths in said computer.

17. The method of claim 12, wherein said computer comprises a personal computer.

19. An article comprising:

a machine-readable storage medium, said storage medium having stored thereon instructions, said instructions, when executed by a computer system including an audio channel, resulting in the following steps:

creating at least two signal streams for a waveform in said audio channel;

detecting the presence of the first and the second signal sample streams for said waveform at a point in said audio channel; and

measuring the time between the detections of the signal sample streams.

20. The article of claim 19, wherein the waveform comprises a chirp waveform.

21. The article of claim 19, wherein the computer system including an audio channel comprises a personal computer system including an audio channel.

22. An article comprising:

a machine-readable storage medium, said storage medium having stored thereon instructions, said instructions, when executed by a computer system including an audio channel, resulting in the following steps:

creating at least two signal waveforms in said audio channel;

detecting the presence of the first and the second waveforms at a point in said audio channel; and

measuring the time between the detections of the waveforms.

23. The article of claim 22, wherein at least one of the waveforms comprises a chirp waveform.

24. The article of claim 22, wherein the computer system including an audio channel comprises a personal computer system including an audio channel.